AN INTEGRATED SYSTEM APPROACH FOR AUTOMATIC SWITCHING RADIO TELEPHONY

Adhi Widjaja Natahartaka

NAVAL POSTGRADUATE SCHOOL

Monterey, California

THESIS

An Integrated System Approach
For Automatic Switching
Radio Telephony

by

Adhi Widjaja Natahartaka

December 1975

Thesis Advisor: G. D. Ewing

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An implementation of I.C.'s in an automatic switching radio telephony will be presented. Components were chosen to be compatible, however, in some part of the circuit where analog and digital signals are present, or signal levels and impedance of the analog signal produce incompatibility, it is understood that interface elements should be introduced.

An Integrated System Approach

For Automatic Switching

Radio Telephony

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Adhi Widjaja Natahartaka
Major, Indonesian Navy

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requirements for the degree of

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An implementation of I.C.'s in an automatic switching radio telephony will be presented. Components were chosen to be compatible, however, in some part of the circuit where analog and digital signals are present, or signal levels and impedance of the analog signal produce incompatibility, it is understood that interface elements should be introduced.

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I. INTRODUCTION

The main idea of this design is to show basic integrated switching circuitry for radio communication that has the capability of substituting for an automatic wire telephony system in some region where it is not feasible to install a wire telephone system and the number of subscribers is relatively small.

The situation described above could be represented for instance by:

1. People living on small islands.
2. People working offshore drilling.
3. Rangers on mountain tops.
4. Emergency services.
5. A new or temporary military base.

The number of channels used in this system were chosen only as a function of the basic requirements that will be shown in the next chapter. A brief discussion about the channel numbers will also be found in the next chapter.

II. SYSTEM DESIGN

A. REQUIREMENTS

1. This system should provide a facility for 50 persons to communicate with each other, within a radial distance of 25 miles.

2. Every person should be able to call or be called.

3. Talking pairs should not be heard or disturbed by other subscribers.

4. Switching procedure should be simple and fully automatic (without an operator).

5. Communication should have the quality of wire telephony or better.

B. THE THEORETICAL CONSIDERATION OF THE PROPERTIES OF THE SYSTEM

1. The Percentage Freedom of Calls

Statistical data should be taken to determine the real need of how much freedom of call should be provided for a system. Depending on the nature and the activity of the persons to whom the system was dedicated, a choice could be made based upon:

The average waiting time for each trial of call.

The percentage of success of each trial of call.

The cost and effectiveness of additional perfection, etc.

For instance, if this system should have ten percent freedom of call, there should be at most five subscribers capable of initializing a call at the same moment.

2. Switching Efficiency

Switching management would be more efficient if it was arranged by one central office, rather than direct connection from one subscriber to another. Each subscriber should direct the call to the central office; the central office would call the desired counterpart. Using this method and having ten percent freedom of call we need 5 x 4 carrier frequencies. Each subscriber should have a receiver capable of receiving 10 different frequencies, and a transmitter capable of transmitting 10 different carriers. The central office should have 10 different transmitters and receivers.

However, this method had little flexibility for an extention, due to technical complexity of the transmitters and receivers. If we had a system of 200 subscribers with 10% freedom of call and every single subscriber was required to have a receiver and a transmitter capable of shifting to 40 different carriers, it would be impractical.

A more efficient plan would be to divide the communication area into smaller regions and assign a number of carriers for each region in accordance with the required degree of freedom of call. By doing this, the number of carriers in each subscriber could be kept to a certain limit while extention still could be performed.

A higher density of switching circuitry was required to implement this. The reason for dividing the communication area into smaller regions was to shift the complexity from the transmitters and receivers into the switching circuitry, which could then be advantageously performed by ICS.

For a 200 subscriber system having 10% freedom of call, the required carriers would be reduced to 8, if the communication area was divided into 5 regions. (Each subscriber should be able to shift to 8 different frequencies.) The higher the number of subscribers, the more division should be made.

3. Electromagnetic Compatibility and Power Efficiency

Without region divisions, the central office should use omni-directional antennas to cover 360 degrees direction. Applying the region division each antenna of the central office was required to furnish only one region, for this a directive antenna would be suitable. This justifies the EMC and power efficiency stated above. Even more, in regions where interference was minimal, the same carrier could be assigned repeatedly.

4. Area Division

The communication area would be divided into 5 regions containing approximately the same number of subscribers. The central office would be placed at the center of the area to allow operating the transmitters with the same power. Figure 1 shows this situation.

For each region three frequencies would be assigned:

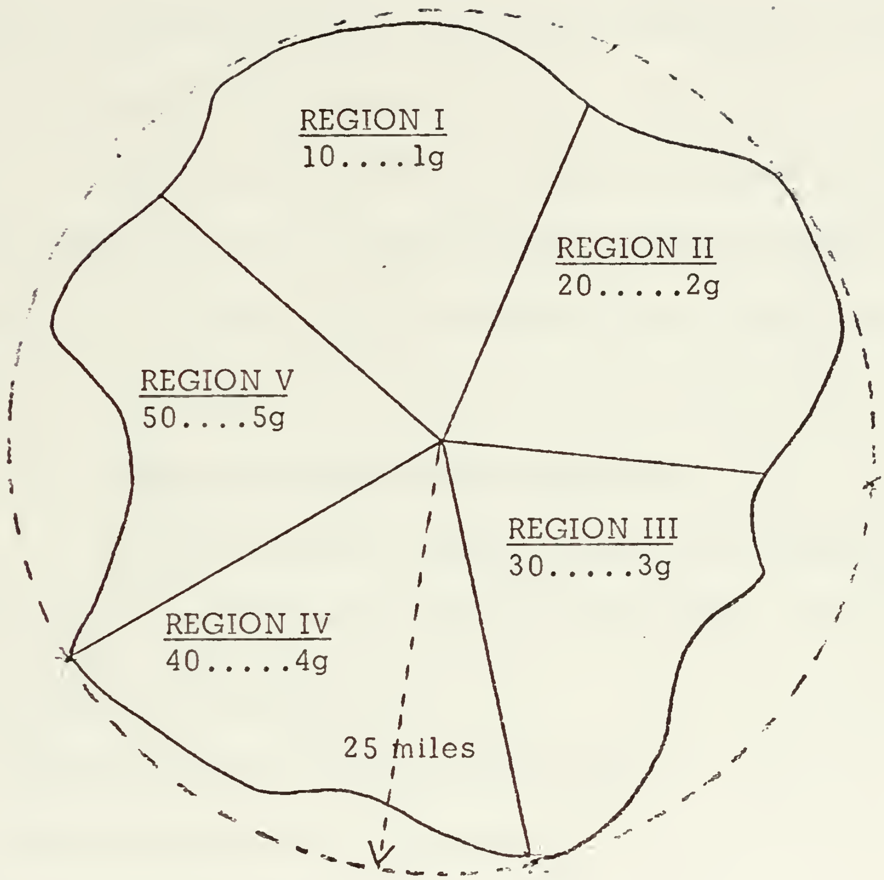

REGION I
10....1g
REGION II
20.....2g
REGION V
50....5g
REGION III
30.....3g
REGION IV
40.....4g
25 miles

Figure 1

The first would be a frequency transmitted by the subscriber and received by the central office.

The second would be a frequency transmitted by the central office and received by the subscriber.

The third would be a frequency transmitted by another subscriber in reply to the calling signal. This is the case where a subscriber was called by someone in the same region.

This method implied that a common frequency be used for the central office to reach the subscribers, but different frequencies were used by the subscribers if there happened to be a talking party in one region.

5. Frequency Allocation and Bandwidth

In brief, the most efficient frequency allocation for this system would be in the higher part of the VHF region. Some reasons that could justify this assumption are:

Suitable for line of sight communication.

Minor changes through the season.

Small fading effect.

Suitable for simple directive antennas.

Less complication compared to the higher spectrum.

Still suitable for small bandwidth.

Reasonable cost for low traffic communication.

The audio bandwidth should be sufficient to support a natural human voice transmission. The standard bandwidth for this purpose would be from 300 Hz to 3200 Hz.

6. Functional Operation

a. Introduction

The design of this system would not take 'freedom of call' into consideration. It was based only upon the requirements that:

each person should be able to call and,

each person should be able to be called.

As a result of this basis, it would be obvious in the coming discussion that:

Five persons could successfully initiate a call simultaneously if they were calling to subscribers in their own regions, or,

Two persons successfully initiate a call to two other regions plus one person call to his own region, simultaneously.

An example of a talking party will be described using a functional block diagram in order to give a clear picture on how the system works. The example will describe how subscriber number 10 was initiating a call to subscriber number 20 (refer to Figure 2).

b. The procedure to initiate a call

Lift telephone handset

Push 'ID' button

Set numbers to be called by pushing suitable buttons

Push 'start' button

After call is over push 'clear' button and return the handset.

c. Lifting up the handset

This action switched on the signaling unit and the transmitter.

d. Pushing ID button

This action sets the ID number into a register, starting a clock and the ID number into a register, starting a clock and sending the numbers in that register in sequential code to the tone signaling unit. (After those pulses modulated the tone carrier they were transmitted.)

These codes would be received by the central office and checked. If this was recognized, it would be transmitted back to the region where it came from. This would be picked up again by that particular subscriber to open his receiving channel. Other transmitters receiving this would switch their transmitters to potentially operating at the second carrier frequency available, but they had not opened their receiving channel yet.

e. Setting up the numbers

This action means preparing another state in the register to be put out next. Subscriber number 20 would be interpreted by the signaling device and the ID detector as: subscriber number 0 in region 2. The sequential code would respectively be the region number followed by the subscriber number.

f. The start button

Pushing this button triggers the clock and starts the transmission of the sequential codes. After receiving ID code from a

subscriber and recognizing it the control unit of the switching circuit in the central office for region 1 would flip from initial state to the second state, ready to receive the region and the subscriber number to be called. This process was indicated by moving the arrow contact from contact point #1 to #2. The region number as well as the subscriber number would be stored in a register. The region part would be examined by the region detector. Utilizing the region number, the region detector would choose the desired direction, namely:

The transmitter of region 1 would be connected to the receiver of region 2, and,

The receiver of region 1 would be connected to the transmitter of region 2.

After connection a clock would be started and the second half of the code (the subscriber number) would sequentially be transmitted heading for the transmitter of region 2. After this the controlling unit switches to the third state. In the third state the receiver would be prepared for receiving voice information, and would directly transmit it via the transmitter of region 2. Meanwhile the subscriber's number (zero) reached region 2, only subscriber number 0 would open its receiving channel and ring a bell. Other subscribers in region 2 changed the carrier frequencies of their transmitters, potentially capable of operating on the second carrier frequency. If subscriber number 0 in region 2 lifted up the hand set, communication between 11 and 20 would be established. From the description above it was also seen that the communicating party

could not be disturbed or be heard by any other subscribers right from the beginning of the signaling states.

g. The clear button

Pushing this button means returning all units to the initial state. This should be done only after the communication is over, otherwise it would disturb the transmission. Returning the hand set cut the transmitter power off (including the signaling unit). In the idle time all the receiver and detection units were on.

7. <u>Codes Rate</u>

The rate at which the codes should be transmitted depend upon some factors:

1. It should not be too slow as to waste too much unnecessary time before a communication could be established.

2. It should not be so fast that it would create a spectrum wider than the available bandwidth.

3. These codes were to be transformed into tone signals and propagated through the voice frequency channels, so that the higher limit would depend upon the voice bandwidth of the communication channels. As was stated earlier it would be between 300 Hz to 3200 Hz.

4. Convolution theorem. Multiplication in time domain = convolution in frequency domain.

From the theory of convolution it was concluded that:

$$F_1 + \frac{1}{\tau} \quad < \quad 3200 \text{ Hz}$$

$$F_1 - \frac{1}{\tau} \quad > \quad 300 \text{ Hz}$$

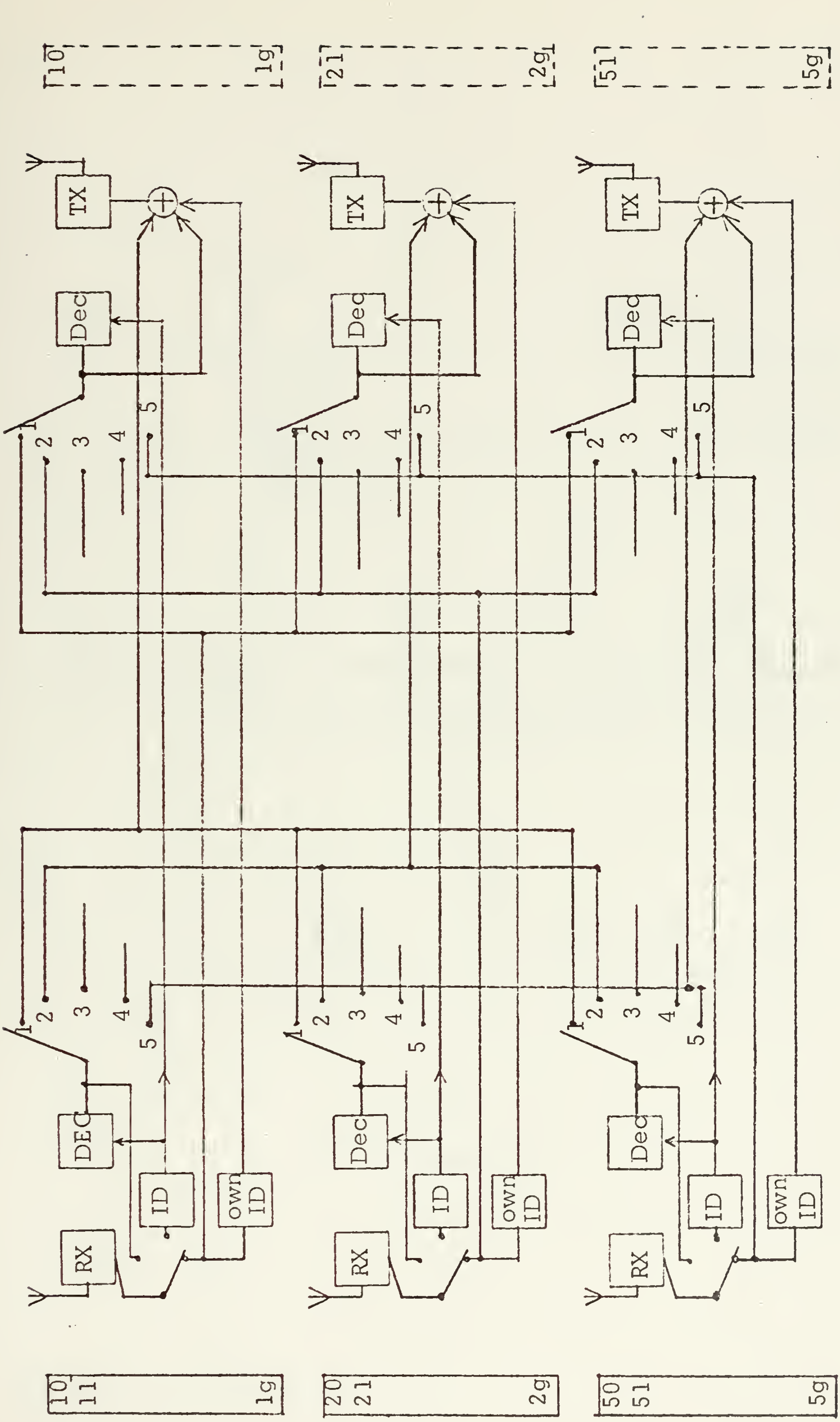


Figure 2

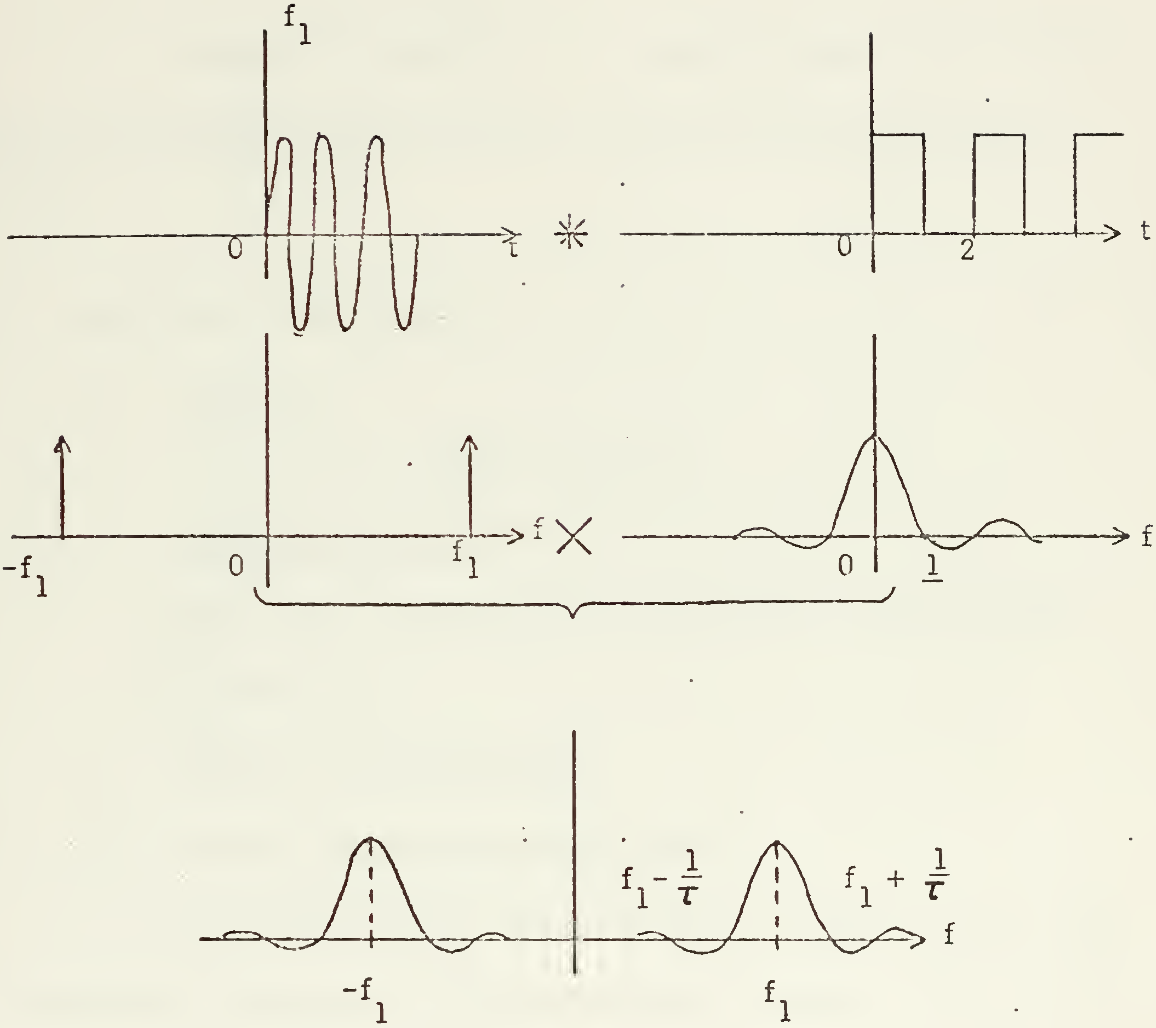


Figure 3

implies:

$$\tau > \frac{1}{3200-f_1}$$

$$\tau > \frac{1}{f_1-300}$$

Both limits show that has only lower limit. So the upper limit depends upon the optimum delay time for the switching purpose.

C. HARDWARE AND LAYOUTS

1. Hardware

Push button switching terminals

Switching system

Solid state, narrow band FM transmitters and receivers in VHF band

Directive arrays antennas

2. Layouts/Placement of the Hardware

Each subscriber should be equipped with one receiver and one transmitter capable of shifting to another frequency. One signaling device, one signal detector and one signal identification device should also be available in order to be able to process the signal. Two directive antennas are used to transmit and to receive signals. For every region, the central office should be equipped with two receivers, one transmitter, one signal detector and identification device, two switching control devices and one matrix switch.

III. JOB DESCRIPTION

1. Design suitable digital codes for signaling purposes.
2. Design a signaling device for the individual subscriber.
3. Design code identification devices for individual subscribers.
4. Design channel openers and carrier selectors for the subscribers.
5. Design code identification devices for the central office.
6. Design a digital region selector for the central office.
7. Design a 5 x 5 matrix switch for the central office.
8. Recommend a suitable communication ratio available from other designers to be used in conjunction with the switching circuitry.
9. Design of suitable antennas.

IV. CIRCUIT REALIZATION

A. THE CODED WORDS

1. Type of Coding

Numbers	Codes
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001
10	1010

No number was coded by all one's or all zero's, these would be used for state indicator.

2. Sub-Carrier Codes

For a 'one' use 2200 Hz.

For a zero use 1800 Hz.

For guard band nothing would be transmitted.

Figure 4 represents a 1001 code word.

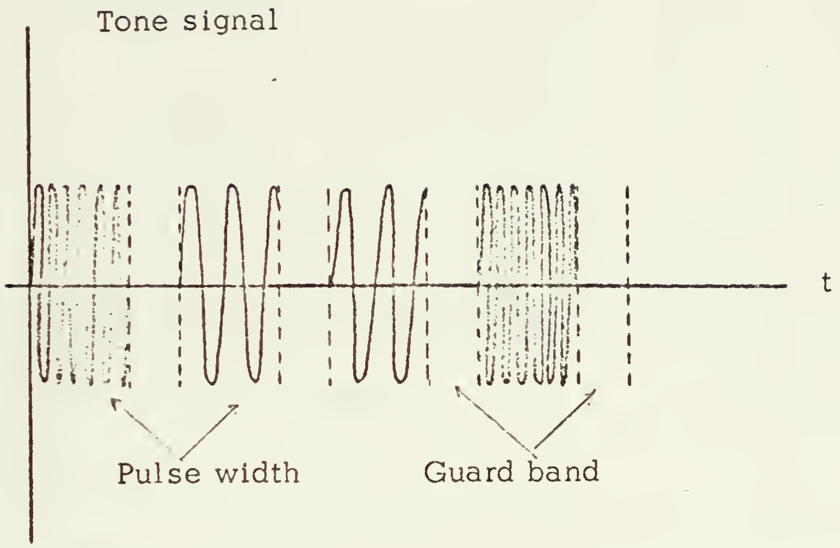


Figure 4

3. Pulse Width and Pulse Period

Applying the formula written before:

$$\tau > \frac{1}{3200-f_1} \qquad \text{the higher sub-carrier}$$

$$\tau > \frac{1}{f_1-300} \qquad \text{the lower sub-carrier}$$

The limiting value of would be:

$$\tau > \frac{1}{3200-2200}$$

$$\tau > \frac{1}{1800-300}$$

1 milli sec

Selected values from the calculation above are:

Pulse width. τ. = 8 milli sec.

PRF. = 100 Hz.

These codes could be received by a filter with one tone circuit of

Q = 18.0

B. THE SIGNALING DEVICE

1. Objectives

This device should perform the coding process of three different signals:

1. The ID numbers.
2. The region numbers.
3. The subscriber numbers.

For different regions different carriers were used. This allows the use of identical codes for subscribers in different regions without getting mixed up one with the other.

The number of subscribers in one region was ten so a 4 bits code word would be sufficient. For uniformity of coding, which would be easier to handle, the region numbers were also represented by a 4 bits coded word (it should be sufficient with three though).

2. <u>Circuit Description</u>

a. Elements of the circuitry

The diagram of this circuit can be seen in Figure 5. The types of the IC elements used for this circuit were:

Shift register 9300

JK flip flop

AND/OR gates

One shot

Pulse counter

Clock generator

The initial condition were all zero's. The clock put out only 8 pulses every time it was triggered. Both ID button and start button would trigger the clock when it was activated. The hand set, when it was lifted up, would perform three functions:

It switched on the signaling unit.

It cut out the bell from its potential power supply.

It switched on the transmitter.

The ID circuit shown in Figure 5 was intended for subscriber #1 (ID = 0001).

b. The ID button was pushed

AND gate 1 was low

Reg. 1 was enabled for parallel input

JK1 was low

JK2 was low

AND gate 3 was prepared for transmission

The logic 1 at the input of P3 of register 1 was transmitted through OR gate 3 to OS1. OS1 produced one clock pulse and transmitted through AND gate 3 - OR gate 1 - CP1. Now the ID code had entered Reg. 1. Note: OS1 was triggered by the leading edge of the ID pulse.

A logic 1 was also transmitted to OS2 and OS3 only when the logic 1 flipped to 0, OS2 and OS3 would be triggered. Note: OS2 and OS3 were triggered by the trailing edge of a pulse.

OS2 transmitted a pulse to JK1 so the P output of JK1 flipped to logic 1. This pulse was transmitted through OR4 to PE of Reg. 1. Now Reg. 1 was disabled for parallel input but enabled for series input.

OS3 transmitted a longer pulse sufficient to allow Reg. 1 to change into a series input state. The trailing edge of OS3 would trigger the clock. JK2 was also triggered by the pulse coming from OS1, so the P output of JK2 flipped to logic 1. Hence: AND gate 2 was prepared for transmission; AND gate 3 was disabled.

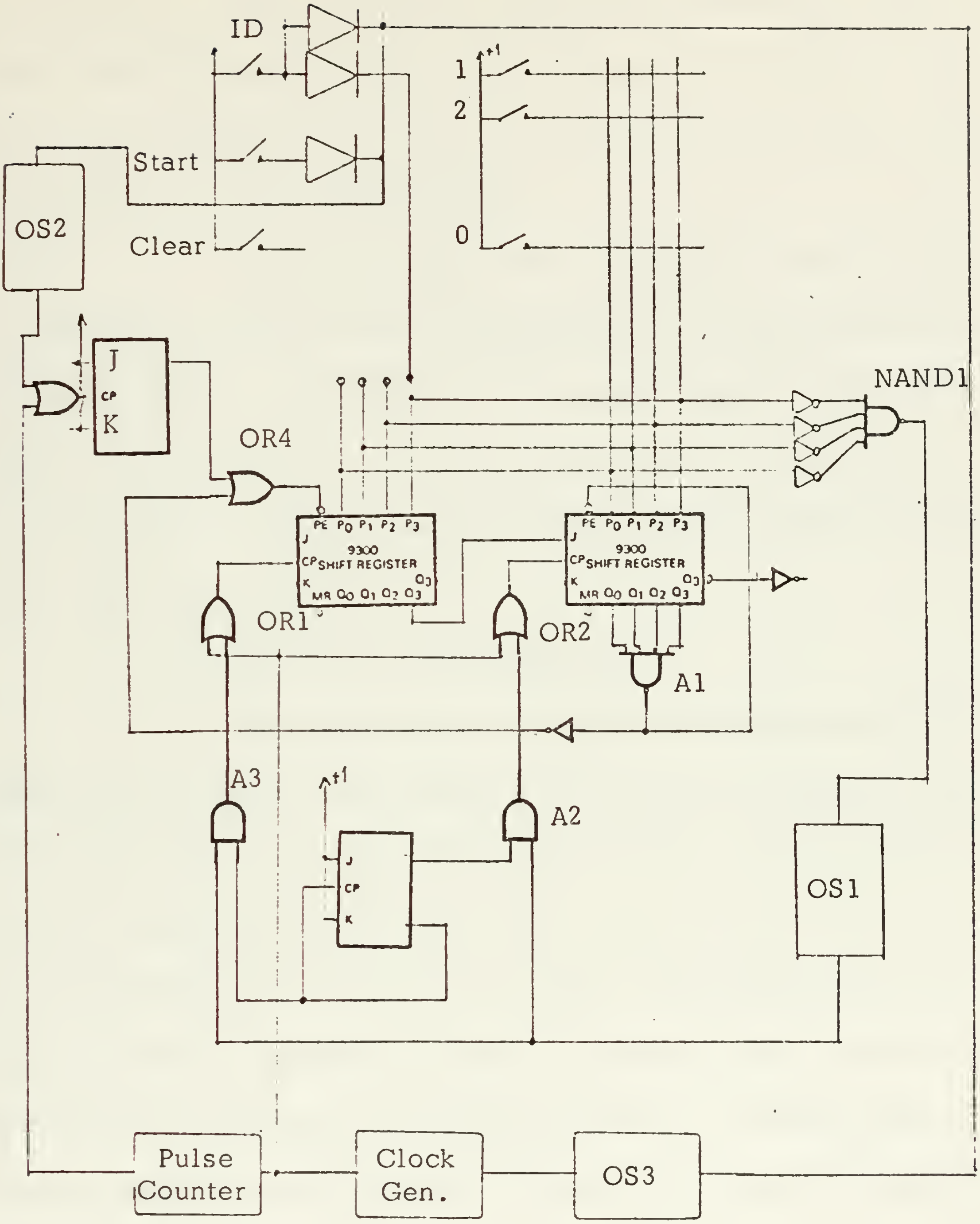

ID
1
2
0
Start
Clear
OS2
J
CP
K
OR4
NAND1
PE P0 P1 P2 P3
J
9300
CP SHIFT REGISTER
K
MR Q0 Q1 Q2 Q3
OR1
OR2
A1
A3
A2
OS1
Pulse Counter
Clock Gen.
OS3

Figure 5

Reg. 1 and Reg. 2 were both in position for series input. When the clock started to pulse they were transmitted through OR1 to Reg. 1; through OR2 to Reg. 2, driving the content of both registers all the way out to the signaling device.

The new contents of Reg. 1 and Reg. 2 were all logic 1. AND gate 1 turned to logic 1, keeping Reg. 1 in series input state, in case JK1 changed state, which it was, since the pulse counter transmitted the last pulse of the clock through AND gate 4 - OR5 to CP of JK1..INT1 was low allowing Reg. 1 for parallel input. It was in the state ready to receive region code.

c. The region number was pushed

The region codes were collected together and transmitted by OR3 to OS1. OS1 produced a pulse and was transmitted through AND gate 2 - OR2 to CP2, clocking register 2 and the region number was received. AND gate 1 was disabled, PE of Reg. 1 turned to low (JK1 still low allowing Reg. 1 to receive parallel input. At the same time INT1 turned to high preventing Reg. 2 from further parallel input. JK2 also received the pulse from OS1 and flipped to logic 0. AND gate 2 was disabled, AND gate 3 was enabled. The state was ready for a subscriber input.

d. The subscriber number was pushed

The pulse was again collected by OR3 and transmitted to OS1. OS1 transmitted a pulse through AND gate 3 - OR1 to CP1, clocking Reg. 1 and the subscriber code was received by Reg. 1. JK2

also received the pulse and was turned down to logic 0 again. Now the state was ready for a start signal.

e. The start button was pushed

This signal was received by OS2 and OS3. OS2 transmitted a pulse and turned JK1 to logic 1. OR4 also turned to logic 1, PE of Reg. 1 turned to high, Reg. 1 was prevented from further parallel input but prepared for series input. The trailing edge of this pulse triggered the clock. The clock pulses were transmitted through OR1 and OR2 to drive the contents of Reg. 1 and Reg. 2 all the way out to the tone signaling device. The signal process was done.

3. <u>A Possibility of Improvements</u>

ID transmission could be performed by the action of lifting the hand set. Hand set switch triggered a one shot and this pulse went to a ROM that had been programmed for an ID code. So the ID button could be eliminated.

Start signal could be performed by collecting the pulses from OS1 in a counter. Through the whole process of code preparation there would only be three pulses transmitted by OS1. If the counter was designed to put out one pulse of the proper length for every three pulses received then transmit this pulse to the start switch. This would perform the starting process. Thus the start button could be eliminated.

The 'clear' process could also be done by the lifting up action of the hand set. The created pulse should precede the ID pulse by a proper amount of time. In fact this method should work better since the

'clear' action was done at the beginning of the whole process, insuring that the initial states were really at logic 0. This fact eliminated the possibility of a faulty initial condition due to a transient produced during the switching on action of a power supply.

Comment:

For a wrong connection the remedy could be to return to hand set and to start the procedure again from the beginning. This would be like the general telephone system.

This information was presented this way in order to simplify the explanation and to emphasize the three important activities that should be performed by this system:

1. The identification.
2. The signaling for the desired subscriber.
3. The returning to the initial state.

C. THE CODE IDENTIFICATION, THE VOICE CHANNEL OPENER AND THE CARRIER

1. Objectives

These devices are to perform:

1. The identification of the incoming signals from the central office.
2. Controlling the voice channel in accordance with the signals received.
3. Controlling the carrier selection.

Here these devices would be presented simultaneously in order to give a more understandable operation in conjunction with one another. The necessary details can be found in Appendix B.

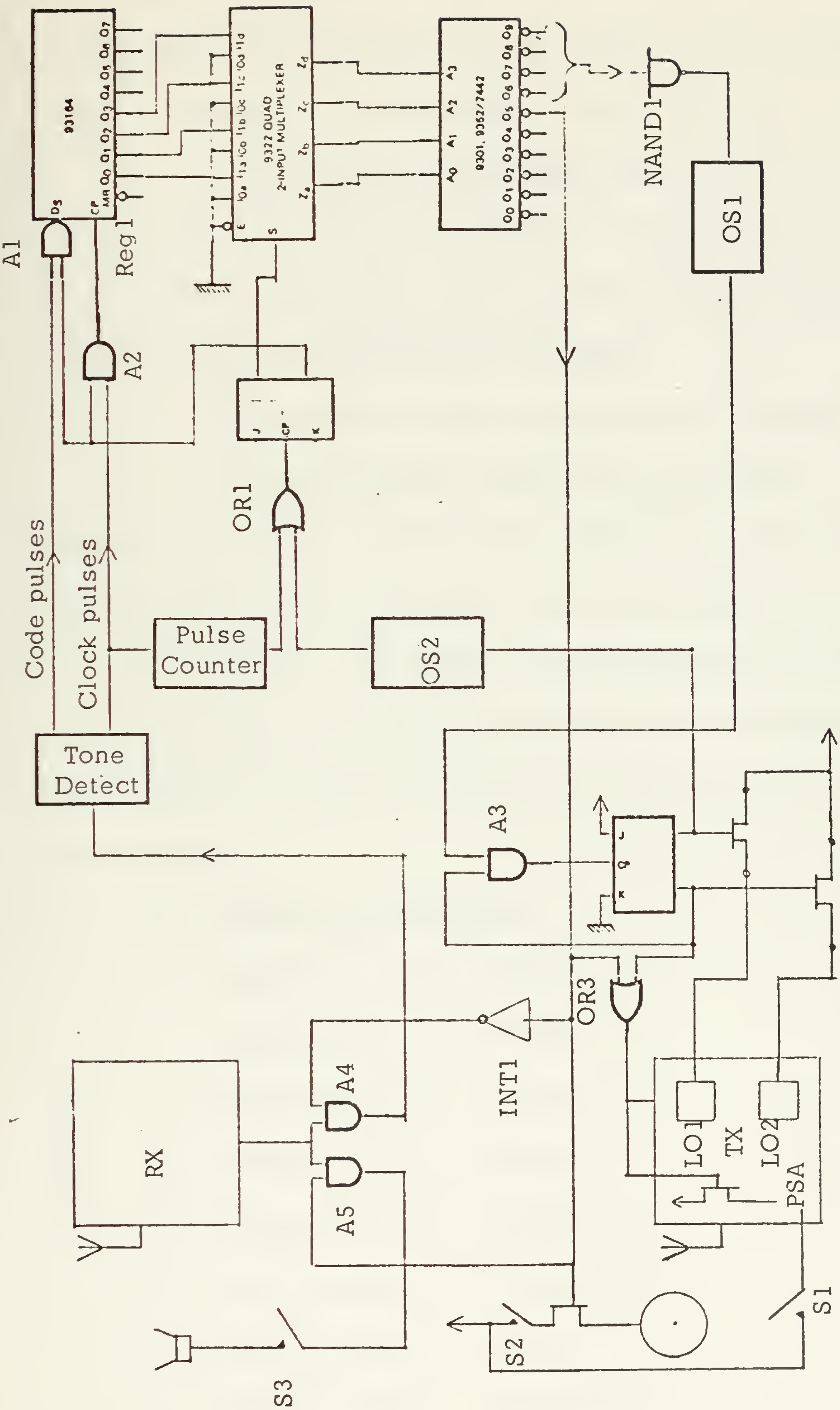
2. The Operation

The block diagram of this circuitry can be seen in Figure 6. There are 5 distinct operations performed by this circuit.

When a particular subscriber initiates a call, the first code he sends is the ID number. This would be received by the central office, checked by the central office and if it was O.k., transmitted back by the central office to the region where it came from. This code would be received again by that particular subscriber and it would be used to open the voice channel.

When someone else calls another person also in his region, two distinct ID's would successively be received by him. (He was not called but two ID's would successively be received by him, the first ID was the initiator's, the second ID was the address, both were from the same region where he was located.)

At the first "ID" it would switch a local oscillator to be prepared for operating on the second carrier frequency available. It also instantaneously shuts off the power supply of the transmitter, thus it would not be possible to be set on by the action of the hand set. He would be kept silent and neither could he listen. The second ID would not affect anything, thus he was still in silent and in a non-listening condition.



A1
Reg 1
9372 QUAD
2-INPUT MULTIPLEXER
9301, 9352/7442
NAND1
OS1
A2
OR1
Code pulses
Clock pulses
Pulse Counter
OS2
Tone Detect
A3
OR3
INT1
A4
A5
RX
LO1
TX
LO2
PSA
S1
S2
S3

Figure 6

When someone else got a call from another region only a single unknown ID was received. This would create a condition which was the same as the first part of the previous situation.

When he was called by somebody in his region, two ID's would be received. The first would be an unknown ID, the second would be his own ID. When the unknown ID was received he would be at the same condition as mentioned before, namely:

Change the potentially operating local oscillator.

Shut off the power supply of the transmitter.

However, when the second ID came, which was his own, the power supply would be reconnected and a bell would be rung.

When he was called by someone from another region, only his own ID would be received, this would create the same condition as he was calling, except the hand set was still on its rest, so the bell would be rung.

3. <u>The Circuit Components</u>

Registers	93164
Multiplexers	9322
JK flip flop	9N73
One shot	9603
Decoders	9301
Tone Detector	Appendix B
Pulse counter	9305
Pulse generator	Appendix C

4. Circuit Description

The initial conditions would be zero states.

JK flip flop #1 was low

JK flip flop #2 was low

AND gate #1 was enabled

AND gate #2 was enabled

AND gate #3 was disabled

AND gate #4 was enabled

AND gate #5 was disabled

AND gate #6 was disabled

The incoming audio pulses would be detected by the tone detector and transformed into two types of DC pulses.

1. The signal pulses. They consist of zero's and one's corresponding to the original codes.

2. The clock pulses. They consist of successive positive pulses each corresponds to either one of the audio tones. (Tone frequency #1 and tone frequency #2 produced the same positive pulse.)

When the first sequence of pulses arrived, those audio signals would go through AND gate #4 to the tone detector. The DC signal pulses from the output of the tone detector would go to AND gate #1 - Reg. #1. The DC clocking pulses would go through AND gate #2 to the clock input of Reg. #1 (CP1). They would also go to the pulse counter, where the last pulse would be transmitted through OR gate #1 to JK #2. The trailing edge of this pulse would trigger JK #2 to high.

After these pulses were received by the Reg. #1, JK #2 turned to high. AND gate #1, AND gate #2 were disabled, AND gate #6 was enabled. The decoder was activated.

If this signal was his own ID, the decoder would enable AND gate #5, and disable AND gate #4, thus the voice channel was opened and the transmitter stayed on the original carrier frequency.

In turn OS #3 would trigger JK #1 and turn it to high. If this was an unknown ID, the decoder would trigger OS #3 (one shot) the P output of JK #1 would turn the LO #1 (local oscillator) on. T would also trigger OS #2 to transmit a pulse through OR gate #1 to JK #2. JK #2 was turned to low again. AND gate #1, AND gate #2 were enabled. The Q output of JK #1 turned the LO #1 off, disabled AND gate #3, and disabled the power supply of the transmitter.

At this state the circuit was prepared for other sequence of pulses.

If the second sequence of pulses was unknown, it would not affect anything since AND gate #3 was disabled by the Q output of JK #1.

If the second sequence was his own ID, the decoder would put out a logic one (high) through OR gate #3 to enable the power supply of the transmitter. It also sends a logic 1 (high) to inverter #1 to disabled AND gate #4, thus blocking any further signal from entering the tone detector circuitry. This logic 1 also enables AND gate #5, providing a path for further signals to reach the output terminal, in this case a loud speaker. The logic 1 also operated the switch of the bell to be rung.

After the communication was over, the carrier was off the air. This would trigger a clearing circuit to transmit a clear pulse which would be received by all circuitry.

D. THE IDENTIFICATION AND REGION SELECTOR OF THE CENTRAL OFFICE

1. Objectives

These devices would perform:

1. The identification of the incoming signal.
2. Selecting the desired region for a legitimate signal.
3. Calling the desired subscriber.

These devices will also be explained simultaneously to gain an understanding of how they are related to one another.

2. Circuit Components

Names	Types
2 Registers	93164
1 Register	93165
3 Multiplexers	9322
3 Decoders	9301
5 JK FF	9LS109
1 Clock Generator	Appendix C
1 Pulse Counter	Appendix C
1 Tone Detector	Appendix C
2 Mono Stable Mults.	XXX
3 NAND Gates	9LS30

1 OR Gate Chip	9LS32
1 AND Gate Chip	XXXX

3. Circuit Description

A block diagram of this circuit can be seen in Figure 7. The detection of ID codes was performed by Reg. #1 and decoder #1. The selection of the proper transmitter in conjunction with the desired region was performed by the first half of Reg. #2 and the first half of decoder #2. The transmission of the numbers of the chosen subscriber was performed by the last half of Reg. #2, the last half of decoder #2 and Reg. #3.

Upon receiving a carrier frequency from a subscriber, a switch was turned on to operate the transmitter. After the carrier was removed this switch would be turned off again. A "clear" pulse would be transmitted to every logic circuitry. As a result of this process, all AND gates except A1, A2, A5 and A7 would be disabled. The clock pulses would come from two sources:

1. The tone detector.
2. The clock generator.

Both sources would generate only 8 pulses every active period. The pulse counter would transmit a single pulse every 8 pulses received.

When the first sequence of pulses arrived, they would enter Reg. #1, via A7 and A2. (A3 and A4 were disabled at this moment) The clock pulses would also reach the counter through A7, OR1. The clock pulses reached Reg. #1 through A7, A1, OR2 gates and the last pulse

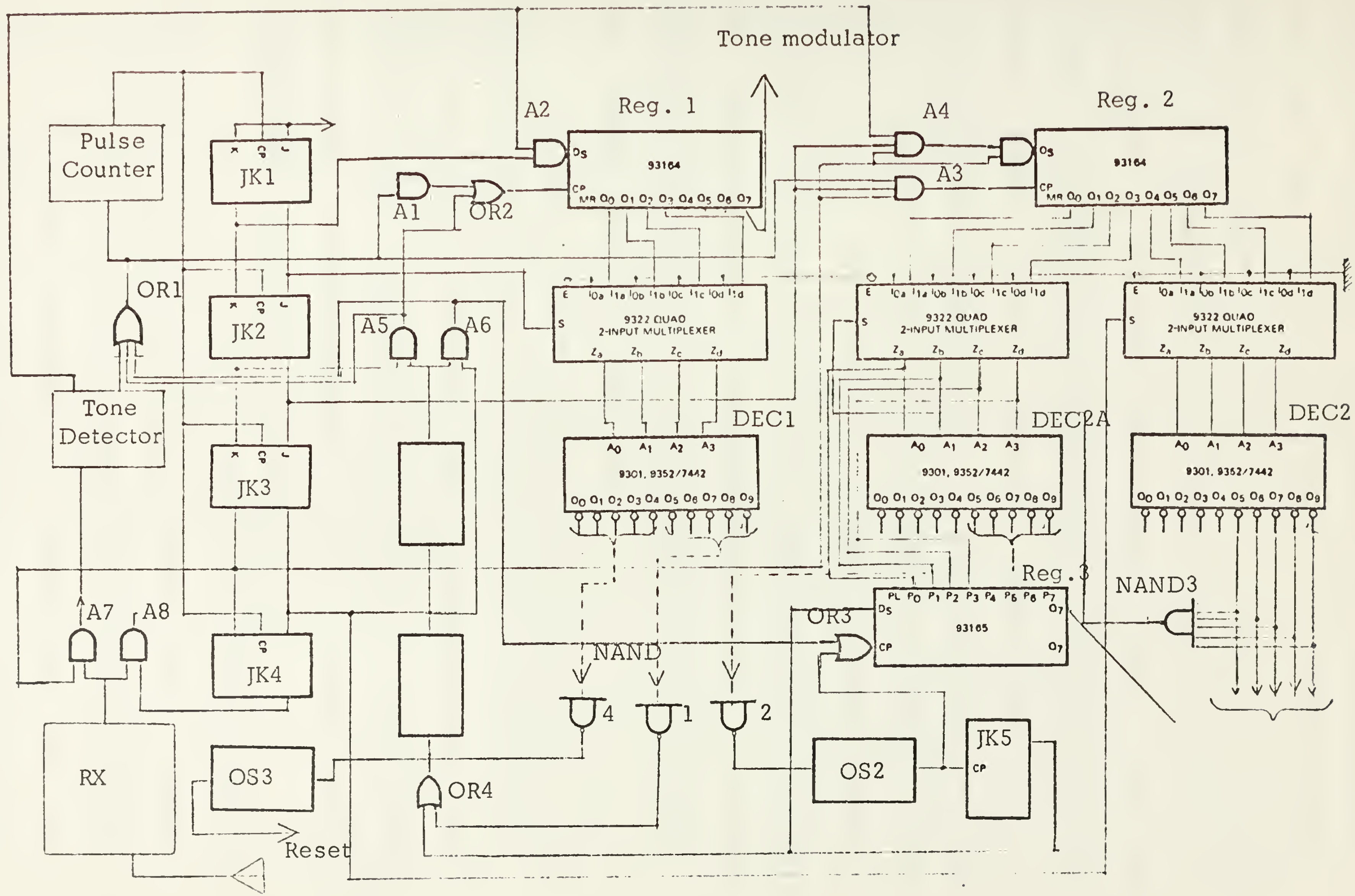

Tone modulator
Reg. 1
Reg. 2
Reg. 3
A1
A2
A3
A4
A5
A6
A7
A8
OR1
OR2
OR3
OR4
Pulse Counter
Tone Detector
JK1
JK2
JK3
JK4
JK5
93164
93164
93165
9322 QUAD 2-INPUT MULTIPLEXER
9322 QUAD 2-INPUT MULTIPLEXER
9322 QUAD 2-INPUT MULTIPLEXER
DEC1
DEC2A
DEC2
9301, 9352/7442
9301, 9352/7442
9301, 9352/7442
NAND
NAND3
4
1
2
OS2
OS3
RX
Reset

Figure 7

would be transmitted to JK1. JK1 flipped to logic 1, the multiplexer 1 was a JK1 flipped to logoc 1, the multiplexer 1 was enabled to select the line input. (Before this the multiplexer was enabled to select ground input.) The signal codes in Reg. #1 were transferred to decoder #1. If this code was recognized by decoder #1, NAND 1 would be enabled and a logic 1 would trigger the clock via OR4-OS1. The clock pulses went through: A5, OR2, to Reg. #1 and drove the contents of Reg. #1 out. The clock would also go through A5, OR1, counter and the last pulse would trigger JK2 to logic 1 (JK1 stayed high). Now A5 was disabled, A3 and A4 were enabled. The logic circuit was in the second state, prepared to receive the next sequence of codes. If the codes were not recognized, the decoder would put out a reset pulse through NAND4, OS2, to all JK flip-flop. This means that the circuit returned to the first state and just neglected the non-legitimate pulse.

When the second sequence of pulses arrived they would go through A7, A4, to Reg. #2. The clock pulses would also arrive at Ref. #2 through A7, OR1 A3 gates. They would also reach the counter through A7, OR1. JK3 would receive a pulse from the counter at the end of eight pulses. JK3 flipped to logic 1, A3, A4, A7, were disabled. This would cause multiplexer #2 to select the line input.

The codes in the first half of Reg. #2 entered decoder #2. One output of decoder #2 went high, and activated one of the matrix cross point switches. This was the process of directing the calling signal to a chosen region by performing a selection of a proper transmitter. The signal also turned the transmitter on. K1 NAND3 put out a

logic 1 that enabled multiplexer # 2A to select the line input. At this moment the contents of the second half of Reg. #2 entered the decoder #1 entered decoder #2A and Reg. #3. Decoder #2, through NAND2, put out a logic 1 to OS2. OS2 transmitted a pulse to Reg. #3. (clocking a parallel input) JK5 also received the pulse from OS2 and flipped to high, preventing Reg. #3 from further parallel input. JK5 triggered OS1 through OR4. OS1 in turn transmitted a single pulse to the clock. The clock transmitted pulses through: A6, OR3 to Reg. #3, so the contents of Reg. #3 were driven all the way out to the selected transmitter. This signal would operate a particular receiver in the chosen region. If the subscriber responded to this call, communication was established.

NOTE: A counterpart matrix crosspoint switch was also designed to choose a receiver assigned to a particular region. This switch was also activated by the same logic level of decoder #2. Thus at the same time a transmitter was connected to a proper receiver and a receiver to a proper transmitter, both were controlled by a logic level of decoder #2. (Recall the functional diagram on Figure 2.) After the communication was over and the RF carrier was off the air a pulse would be transmitted by OS3 to clear all logic circuits. This would disconnect the switches and return to the initial state.

E. THE MATRIX SWITCHES

1. Objectives

The switches should be able to perform:

1. Switching on the proper transmitter.

2. Connecting the calling receiver to the selected transmitter.

3. Inhibit other receivers from selecting the same transmitter.

In order to perform these functions, each switch would physically have to have two contacts. The following circuit description would explain:

The group of matrix switches functioning as channel selector.

The relation between two switches to perform these functions.

2. <u>Circuit Description</u>

a. Matrix switches as channel selectors

The functional diagram of these switches are shown in Figure 8. Each input line can select any output channel. The selection process was controlled by decoder #2 assigned for a particular region. The parallel switches of a given output could only be selected by a single input channel at one time. Circuitry was designed to prevent multiple selection of one output by several inputs at one time.

b. The inhibit function circuitry

Using a pair of switches the three functions of the switching circuit could be performed. Figure 9 shows the functional diagram.

S1 and S'1 were the pair switches. S2 was a multiplexer. When decoder #2 selected S1 and S'1, a current would flow through: R - S'1 - S1 - S2 - to ground. Point "A" would drop to zero.

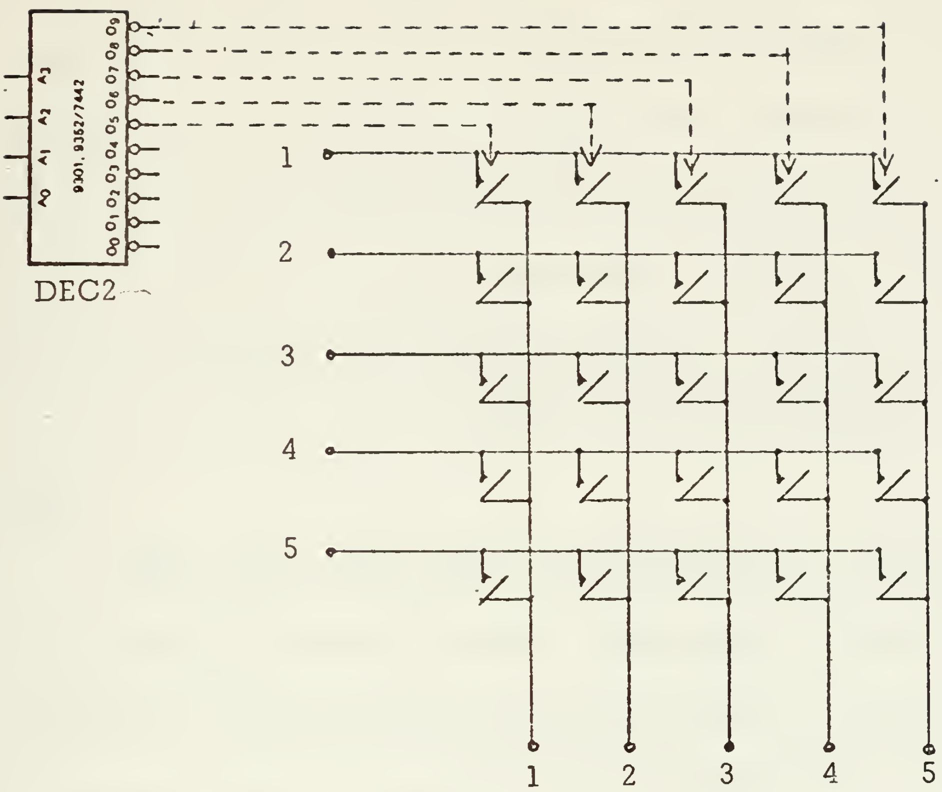


Figure 8

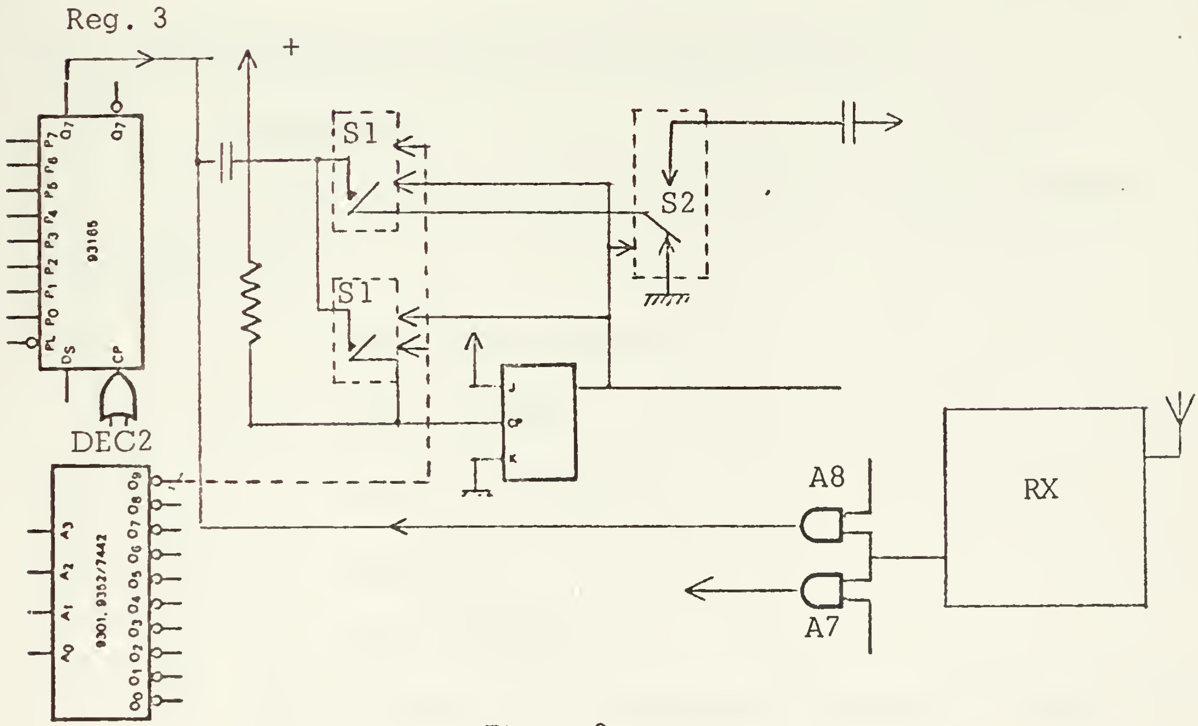


Figure 9

This would trigger the JK flip flop to jump to the next state. S2 was activated by the JK flip flop and connected the channel. The JK flip flop replaces the function of the decoder to close switches S1 and S'1. Any other decoder from another region might transmit a pulse to switches S1 and S'1 but there was no more ground path provided to trigger his own JK flip flop to maintain the switch connection. This action could be seen on Figure 9, shown by the position of switch S2 after it has been activated once.

NOTE: Each output terminal of a decoder had a JK flip flop to complement the switching process. Both were connected to a busy tone generator. The tone generator would work only if it was triggered by the decoder but was not compelmented by the JK flip flop. This was the case where a second input was trying to choose the same output channel.

F. THE TRANSMITTING AND RECEIVING ANTENNAS

1. Introduction

For this system the author prefers to use a YAG1 antenna. The reasons are:

1. Reasonable directivity.
2. Easy to design.
3. Easy to implement.
4. Low Cost.
5. Easy to install.

These features are very important since they are being installed on a special building (the central office) as well as on residential houses.

Directivity is desirable mainly because of the conservation of the output power. It is also true that it helps to control interference among systems, and reduces the picking up of local noise by the receiver.

2. Design Consideration

Using the experimental data for a three element array given on page 25-4 of the Reference Data for Radio Engineers, Fifth Edition, 1973, the geometry of the antenna was designed as follows:

Dipole element: 0.5

Reflector element: 0.5+10% of

Directors elements: 0.5-10% of , and

Successively trimmed by 10% from the previous one for every additional director.

Distance from the reflector to the active dipole is 0.25

The distance from the first director to the active dipole is 0.20 . For every additional director an equal increment of distance should be taken.

3. The Simulation of the Antenna on the Computer

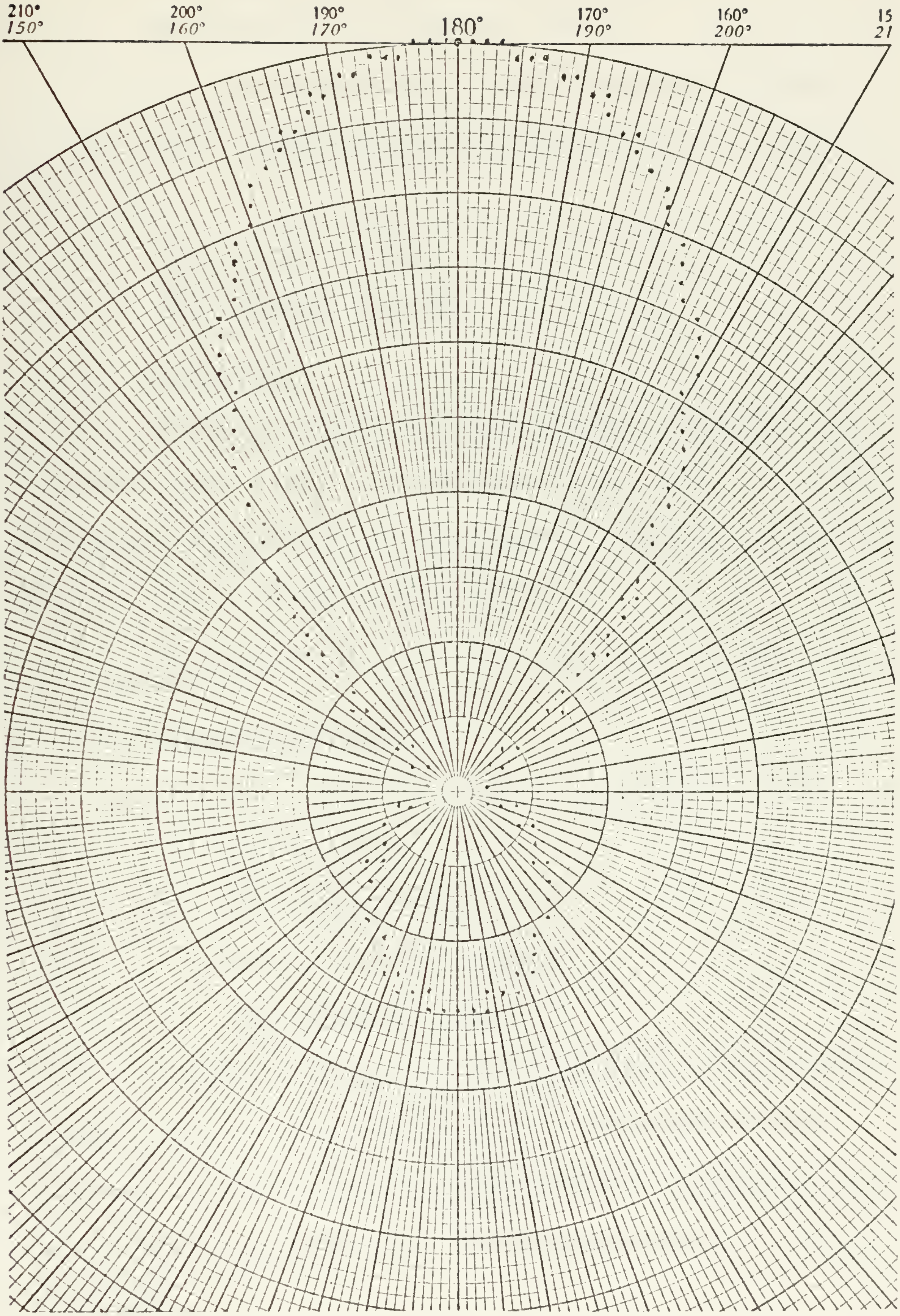
Using the data from the previous page, two antennas were simulated uilizing the 'ASAP' subroutine.

Active element was a folded dipole.

Antenna A had 4 directors and 1 reflector.

Antenna B had 11 directors and 1 reflector.

The length of the active element was: 0.5


210°
150°
200°
160°
190°
170°
180°
170°
190°
160°
200°
15
21

The width of the folded section was: 0.05

The diameter of the wire was: 0.005

The results of both antennas are shown on the following page. On these patterns the only noticeable difference is the front to back ratio factors.

For antenna A: 9.12 DB

For antenna B: 10.46 DB.

4. Conclusion

Based upon the result of the simulations, the author concludes that it will not be of much interest to further increase the number of elements (directors) beyond 11 for this particular type of antenna. The choice is left to the individual taste of the user.

G. THE RECEIVER

1. Noise Consideration

Four dominant sources of noise in the VHF region are:

The man-made noise.

The galactic noise.

The receiver internal noise.

The atmospheric noise.

To give a first approximation of the usable sensitivity of a receiver the noise level presented to the receiver whould be known. Based upon the data presented in the ITT hand book the following figures are considered to be sufficient for representing noist level at a frequency of 150 MHZ.

Man-made noise: 20 DB.

Galactic noise I DBM.

Solar noise: undetectable.

Atmospheric noise: undetectable.

These figures are DB measured above average temperature noise:

$$N = KTB.$$

where at room temperature the value of T was assumed to be 300 degrees. For a receiver bandwidth of 3000 HZ, the expected noise power would be:

$$N = 1.38 \times 10\ (-23) \times 300 \times 3000 \text{ watts}$$

$$1.242 \times 10\ (-17) \text{ watts or}$$

$$N = -169.06 \text{ DB watts.}$$

The expected man-made noise power would be: (20-169.06) DB = -149.06 DB or

$$1.24 \times 10\ (-15) \text{ watts}$$

The expected galactic noise power would be:

$$(2-169.06) \text{ DB} = -167.06 \text{ DB or}$$

$$1.97 \times 10\ (-17) \text{ watts.}$$

By an assumption that those noise powers are additive, the total noise power on a receiver at 150 HHZ would be:

$$1.24\text{x}10(-17) + 1.24\text{x}10(-15) +$$

$$+ 1.24\text{x}10(-15) \quad \text{watts} = 1.272\text{x}10(-15) \text{ watts}$$

$$= -148.95 \text{ DB watts.}$$

2. Receiver Sensitivity

From the previous calculation of the expected noise power presented on the receiver, the usable sensitivity of the receiver would be about -140 DBW.

H. THE TRANSMITTER

1. Introduction

The transmitter was intended to be capable of sending radio signals to a distance of 25 miles. Using the value of the total noise power - 148.95 DBW and considering that a signal to noise ratio of 20 DB would be sufficient for a frequency modulated receiver, the transmitter should have a power output capable of supplying - 128.95 DBW at the receiver.

For a measure of the transmission reliability a suggestion was made by ITT that an additional power over the free space calculation would provide the following results:

10 DB for 90% reliability.

20 DB for 99% reliability.

30 DB for 99.9% reliability.

40 DB for 99.99% reliability.

A 99.99% reliability would be considered necessary for this system because of the following reasons:

The switching codes consist of 8 bits word without check bit so that it needs a high reliable transmission system.

The voice information should have the quality of a wire telephone system which has a low noise level.

2. Power Calculation

P(trans) = N(sig) + A(free) + A(ATM) - G(ant).

where

P(trans) is the transmitted power at the transmitting antenna.

N(sig) is the signal power at the receiving antenna.

A(free) is the free space transmission attenuation. It was calculated using a nomogram for free space path attenuation, presented on page 26 - 20 of the ITT hand book. For a distance of 25 miles, at a frequency 150 MHZ, the attenuation was 104.5 DB.

A(ATM) is the atmospheric attenuation. It was calculated using the atmospheric absorption chart presented on page 26 - 18 of the ITT hand book. The value was found to be: 0.025 DB.

G(ant) is the gain of the antenna with respect to a isotropic radiator.

It was found to be 7.78 DB.

Using these values the expected output power at the transmitting antenna should be -39.985 DBW.

APPENDIX A

THE TONE GENERATOR

OBJECTIVES:

This device should perform the transformation of DC pulses to tone pulses. The positive pulses would be transformed into 2200 HZ tone signals, and the negative pulses would be transformed into 1800 HZ. Using this device the coded words would be transmitted as binary tone signals. Figure 5A shows the circuit diagram of this device and the relation of the individual pulses.

CIRCUIT DESCRIPTION:

When the output of register #2 was high, A2 was enabled, A1 was disabled. At this instance if a positive pulse came from the clock, 2200 HZ (F2) would be transmitted. On the other hand, when the output of decoder #2 was low, a pulse from the clock would activate the 1800 HZ tone generator (F1). Figure 5A(2) shows the clock pulses, it was assumed to start at time T = 0. Each clock pulse was 10 milli sec. long and the pulse width was 20 milli sec. Figure 5A(3) shows the states of the output of register #2, changing with every end of the clock pulses. It was assumed that the train of states in register #2 was: 010101. Figure 5A(4) shows the output of the tone signaling device. It produced tone pulses, changing in frequencies corresponding to the state of register #2. The pulse width was controlled by the clock pulses.

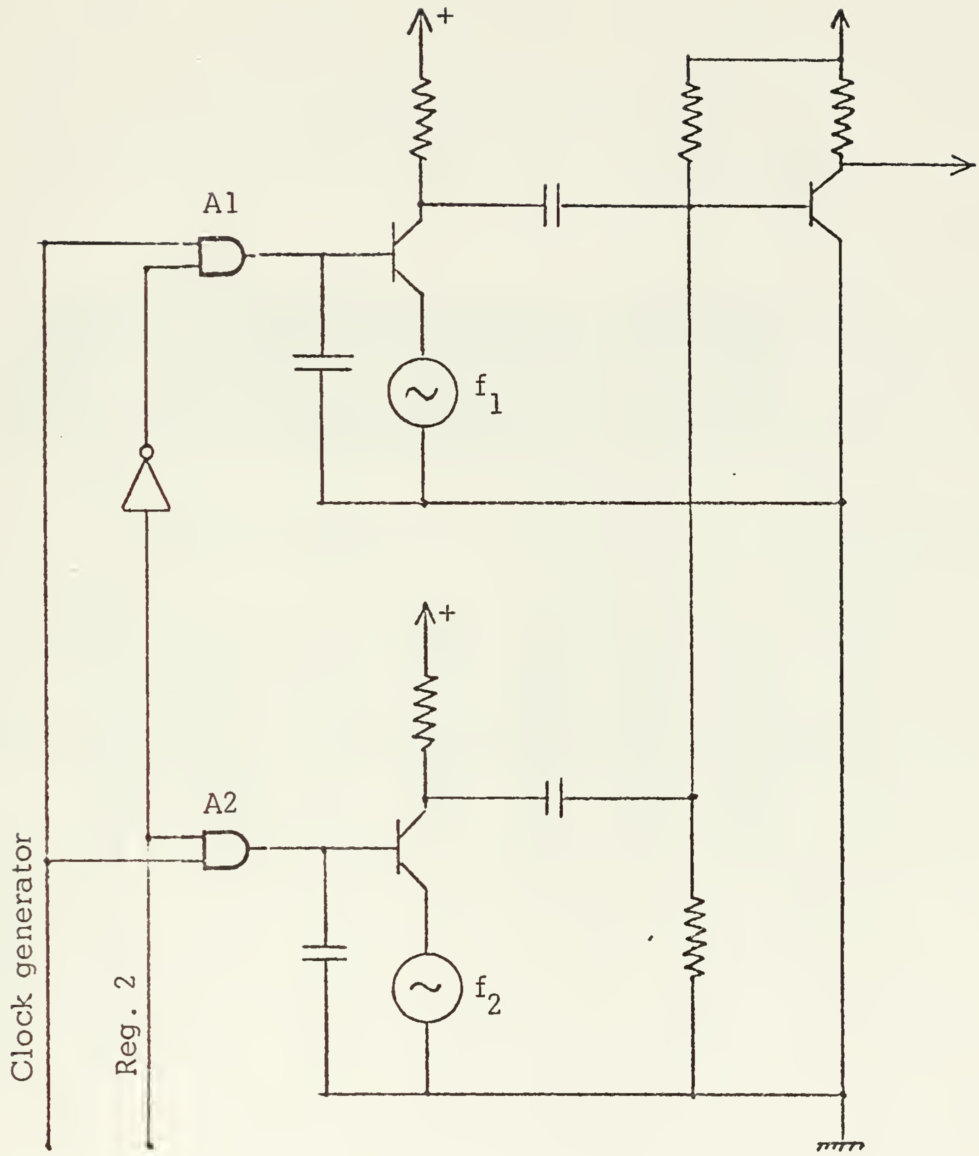


Figure 10

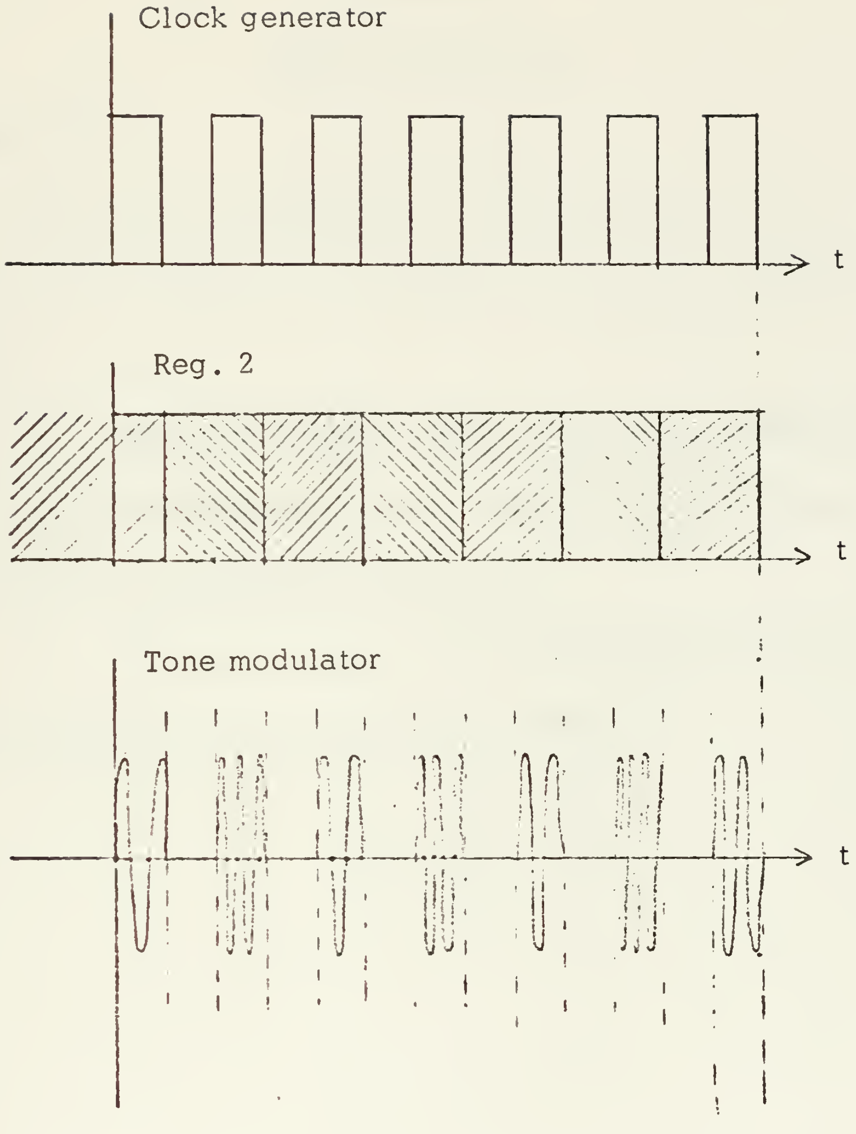


Figure 11

APPENDIX B

THE TONE DETECTOR

OBJECTIVES:

This device should perform the detransformation from voice (tone) pulses into DC pulses. Two types of DC pulses were generated by this device.

1. The clock pulses. These pulses were generated by just rectifying the tone signal pulses. They will have the same pulse width and period as the original clock pulses.

2. The signal pulses were generated by filtering the F2 pulses and rectifying them into positive DC pulses.

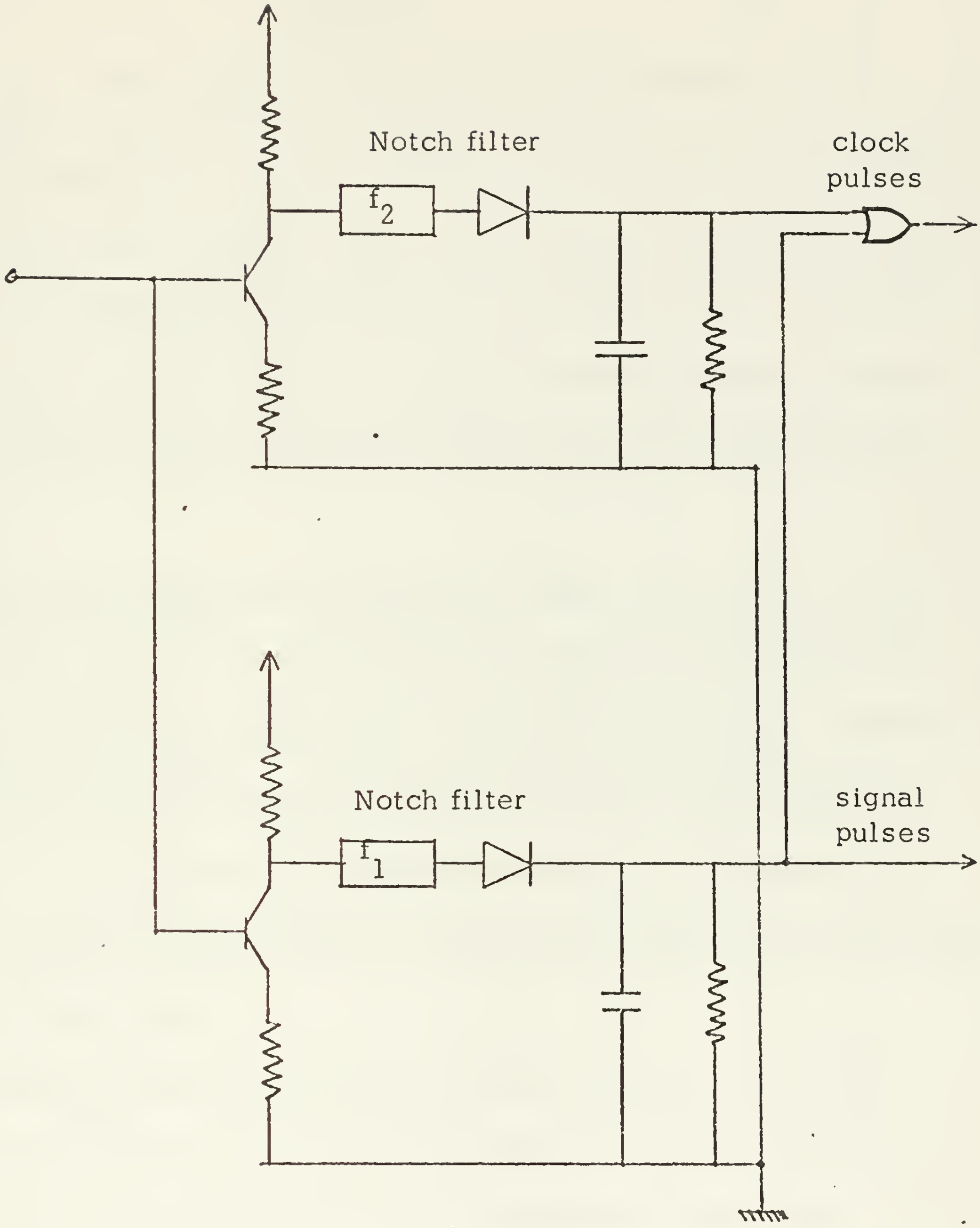


Figure 12

APPENDIX C

THE TONE GENERATOR AND THE PULSE COUNTER

The upper half of Figure 13 shows the pulse generator unit and the lower half the pulse counter. When a pulse came to OR1, JK was triggered to high. The QNOT output of the JK flip flop was low, the clock generator was enabled to put out pulses. The output of the clock always started from low to high and rest on the low logic again after 8 pulses had been sent out.

The clock output also fed the pulse counter through OR2. Using the configuration shown on Figure 13 after 8 pulses had been received, the three output lines of the counter would turn to zero again. Implying that the output of the OR3 gate changed from high to low and OS1 was triggered, a short pulse would be transmitted by OS1. This pulse triggered the JK flip flop and turned it to low again. The clock was disabled from any further output.

The pulse transmitted by OS1 was short enough to prevent the generator from putting out more than exactly 8 pulses. The output of the OR3 gate changed from low to high at the first incoming pulse. This change would not trigger the OS1 since it responded only to the high to low changes.

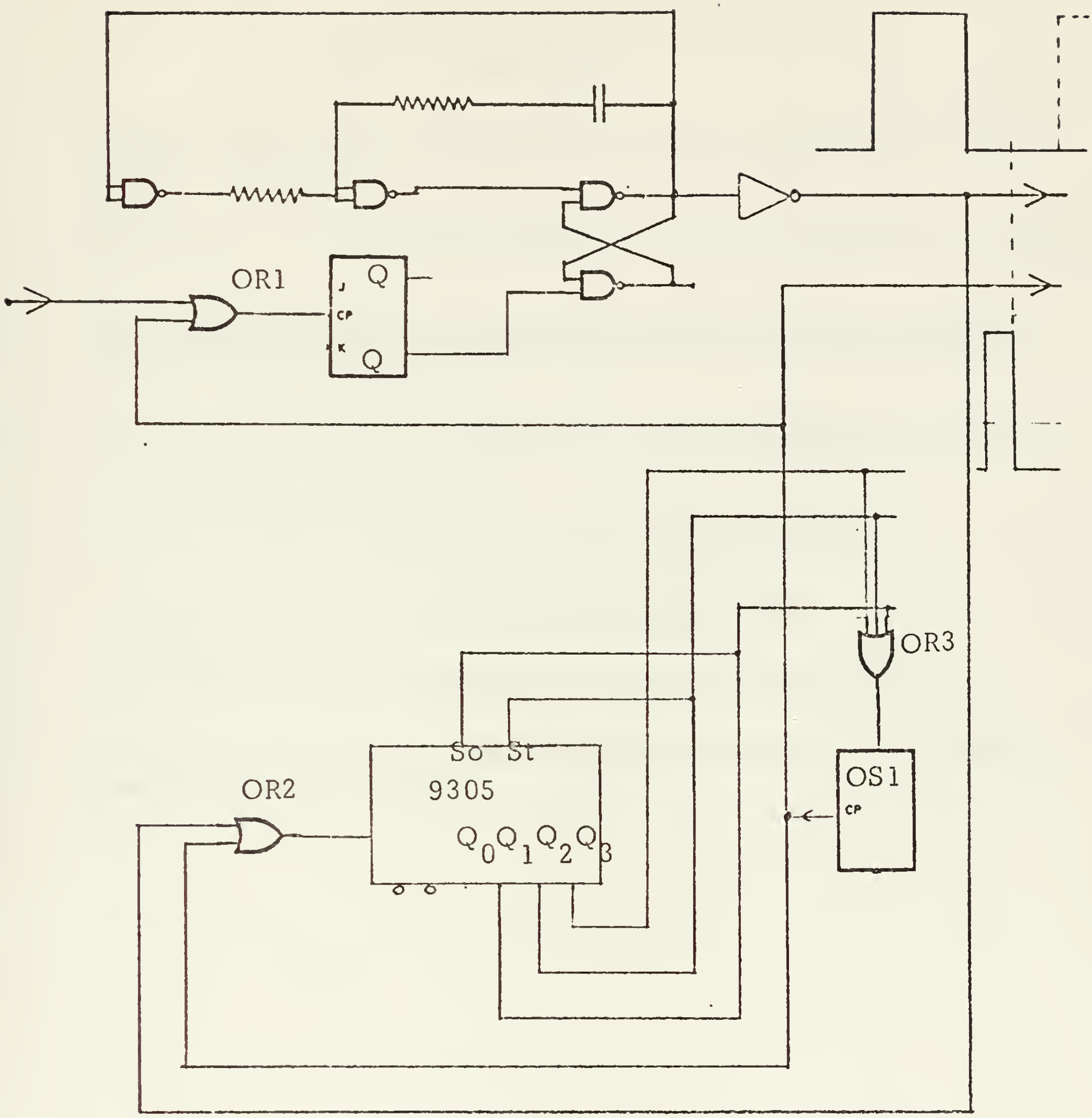


Figure 13

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